AMENDMENTS TO THE CLAIMS

Claims 1, 3-12, 26, 38-49, 74, 91 and 175-192 are pending in the Application. Claims 179-182 were allowed, claims 1, 3-6, 9-11, 26, 38-43, 45-49, 74, 91, 175-178 and 183-192 were rejected, and claims 7, 8, 12 and 44 were objected to in the Office action mailed July 25, 2006. Claims 1, 26, 49, 74, 175, 178, 184, 187 and 190 are amended in this response. Claims 1, 26, 49, 74, 175 and 178-182 are independent claims. Claims 3-12 and 183-185, claims 38-48 and 186, claims 187 and 188, claims 91 and 189, claims 176, 177, 190 and 191, and claim 192 depend either directly or indirectly from independent claims 1, 26, 49, 74, 175 and 178, respectively.

The following listing of claims replace all prior versions, and listings, of claims in the application:

Listing of Claims:

Claim 1. (Currently amended) A signal processing system, for interfacing telephony devices with packet-based networks, the system comprising:

a voice exchange for exchanging voice signals between a network line and a packet based network;

a full duplex data exchange for exchanging data signals from the network line with data signals from the packet based network, wherein the full duplex data exchange demodulates the data signals from the network line, outputs the demodulated data signals to the packet based network, remodulates demodulated data signals from the packet based network, and outputs the remodulated data signals to the network line; and

a resource monitor that monitors processor resources used by one or both of the voice exchange and the data exchange, and that dynamically enables and disables signal processing functionality used by the one or both of the voice exchange and the data exchange <u>in the</u>

exchange of one or both of the voice and data signals of a call, to control processor computational load.

Claim 2. (Canceled).

Claim 3. (Previously Presented) The signal processing system of claim 1 wherein the data signals from the network line are modulated by a voiceband carrier, and the data exchange comprises a data pump for demodulating the data signals from the network line for transmission on the packet based network and remodulating the data signals from the packet based network with the voiceband carrier for transmission on the network line.

Claim 4. (Previously Presented) The signal processing system of claim 3 wherein the data exchange comprises a jitter buffer for receiving packets of the data signals of varying delay from the packet based network and compensating for the delay variation of the data signal packets.

Claim 5. (Original) The signal processing system of claim 4 wherein the jitter buffer outputs an isochronous stream of the received data signals.

Claim 6. (Original) The signal processing system of claim 4 wherein the data pump transmits the received data signals to the network line at a transmit rate.

Claim 7. (Original) The signal processing system of claim 6 wherein the jitter buffer compensates for the delay variation of the data signal packets by holding a number of the received data signals, and wherein the data exchange further comprises a clock synchronizer which adaptively adjusts the transmit rate of the data pump in response the number of the received data signals in the jitter buffer.

Claim 8. (Original) The signal processing system of claim 6 wherein the jitter buffer compensates for the delay variation of the data signal packets by holding a number of the received data signals, and wherein the data exchange further comprises spoof logic which provides spoof data to the data pump when the number of the received data signals held in the jitter buffer is below a threshold.

Claim 9. (Previously Presented) The signal processing system of claim 1 wherein the voice exchange comprises a jitter buffer for receiving packets of the voice signals of varying delay from the packet based network and compensating for the delay variation of the voice signal packets.

Claim 10. (Original) The signal processing system of claim 9 wherein the jitter buffer outputs an isochronous stream of the received voice signals.

Claim 11. (Original) The signal processing system of claim 9 wherein the jitter buffer comprises a voice queue which buffers the received voice signals for a holding time, and a voice synchronizer which adaptively adjusts the holding time of the voice queue.

Claim 12. (Previously Presented) The signal processing system of claim 11 further comprising a tone exchange for exchanging DTMF signals between the network line and the packet based network, the DTMF exchange comprising a DTMF queue for buffering packets of the DTMF signals from the packet based network, and a tone generator which generates a DTMF tone responsive to the buffered DTMF signals, the DTMF queue outputting a signal to the voice synchronizer to suppress the buffered voice signals when the DTMF signals are in the DTMF queue.

Claims 13-25. (Canceled).

Claim 26. (Currently amended) A signal processing system, comprising:

a voice exchange for exchanging voice signals between a first telephony device and a packet based network;

a full duplex data exchange for exchanging data signals from a second telephony device with demodulated data signals from the packet based network, wherein the full duplex data exchange demodulates data signals from the second telephony device, outputs the demodulated data signals to the packet based network, remodulates demodulated data signals from the packet based network, and outputs the remodulated data signals to the second telephony device;

a ~~human voice detector that detects human voice based on~~ pitch period estimator for estimating a pitch period of a voice band signal from one or both of the first and second telephony devices, using an autocorrelation function;

signal power measurement circuitry for producing at least one measurement of power of the voiceband signal from one or both of the first and second telephony devices; and

a call discriminator for selectively enabling at least one of the voice exchange and the data exchange based at least upon a comparison of the pitch period estimate and a plurality of thresholds, and the at least one measurement of power of the voiceband signal ~~the detection of human voice~~.

Claims 27 – 37. (Canceled).

Claim 38. (Previously Presented) The signal processing system of claim 26 wherein the voice exchange comprises a voice decoder for decoding packets of the voice signals from the packet based network for transmission to the first telephony device, a voice activity detector which detects the voice signals without speech, and a comfort noise generator which inserts comfort noise in place of the voice signals without speech.

Claim 39. (Original) The signal processing system of claim 38 wherein the voice exchange further comprises a comfort noise estimator which generates comfort noise parameters from at least a portion of the voice signals without speech, the comfort noise generator being responsive to the comfort noise parameters.

Claim 40. (Previously Presented) The signal processing system of claim 26 wherein the voice exchange comprises a voice decoder for decoding packets of the voice signals from the packet based network for transmission to the first telephony device, a voice activity detector which detects lost voice signals, and a lost packet recovery engine which processes the voice signals to compensate for the lost voice signals.

Claim 41. (Previously Presented) The signal processing system of claim 26 wherein the voice exchange comprises a voice encoder for encoding the voice signals from the first telephony device for transmission on the packet based network, and a voice activity detector which suppresses the voice signals without speech.

Claim 42. (Original) The signal processing system of claim 41 wherein the voice exchange further comprises a comfort noise estimator which generates comfort noise parameters when the voice activity detector suppresses the voice signals without speech.

Claim 43. (Previously Presented) The signal processing system of claim 26 wherein the voice exchange further comprises a decoder for decoding packets of the voice signals from the packet based network, and an echo canceller for canceling decoded voice signal echoes on incoming voice signals from the first telephony device.

Claim 44. (Original) The signal processing system of claim 43 wherein the voice exchange further comprises a non-linear processor which mutes the incoming voice signals when the incoming voice signals do not comprise speech and the echo canceller detects the decoded voice signals with speech.

Claim 45. (Previously Presented) The signal processing system of claim 26 wherein the voice exchange comprises a voice encoder for encoding the voice signals from the first telephony device into voice signal packets for the packet based network.

Claim 46. (Previously presented) The signal processing system of claim 45 further comprising a tone exchange comprising a DTMF detector for detecting a DTMF signal from the first telephony device and generating a DTMF packet for the packet based network in response to the DTMF signal, the DTMF detector muting the voice signal packets when the DTMF signal is detected.

Claim 47. (Previously Presented) The signal processing system of claim 26 further comprising a fax exchange for exchanging fax signals from a third telephony device with demodulated fax signals from the packet based network, wherein the call discriminator selectively enables the fax exchange.

Claim 48. (Previously Presented) The signal processing system of claim 47 wherein the fax. signals from the third telephony device are modulated by a voiceband carrier, and the fax exchange comprises a data pump for demodulating the fax signals from the third telephony device for transmission on the packet based network, and remodulating the demodulated fax signals from the packet based network with the voiceband carrier for transmission to the third telephony.

Claim 49. (Currently amended) A method of processing signals, comprising:

exchanging voice signals between a first network line and a packet based network;

demodulating data signals from a second network line for inputting to the packet based network;

remodulating demodulated data signals from the packet based network for inputting to the second network. line;

simultaneously exchanging the demodulated data signals from the network line with remodulated data signals from the packet based network; and

dynamically enabling and disabling signal processing functionality <u>used in the exchange of one or both of the voice and data signals of a call,</u> to control processor computational load.

Claims 50-73. (Canceled).

Claim 74. (Currently amended) A method of processing signals, comprising:

exchanging voice signals between a first telephony device and a packet based network;

demodulating data signals from the first telephony device for inputting to the packet based network;

remodulating demodulated data signals from the packet based network;

simultaneously exchanging demodulated data signals from a second telephony device with remodulated data signals from the packet based network;

estimating a ~~detecting human voice or lack thereof based on~~ pitch period of a voice band signal from one or both of the first and second telephony devices, using an autocorrelation function;

discriminating between voice signals and data signals based on a comparison of the estimated pitch period and a plurality of thresholds, and at least one power measurement of the voice band signal ~~detection~~; and

invoking at least one of the voice exchange and the data exchange based on said discrimination.

Claims 75-90. (Canceled).

Claim 91. (Previously Presented) The method of claim 74 further comprising:

exchanging fax signals from a third telephony device with demodulated fax signals from the packet based network, wherein the discriminating comprises selectively invoking the fax exchange, and wherein the fax signals from the third telephony device are modulated by a voiceband carrier, and the fax exchange comprises a data pump for demodulating the fax signals from the third telephony device for transmission on the packet based network, and remodulating the fax signals from the packet based network with the voiceband carrier for transmission to the third telephony device.

Claims 92-174. (Canceled).

Claim 175. (Currently amended) A method for interfacing a plurality of telephony devices with a packet based network, the packet based network adapted for transmission of packetized signals, the method comprising:

depacketizing an incoming packetized signal from the packet based network, the depacketized signal having an associated type;

identifying the type of the depacketized signal as one of [[a]] voice signal, [[a]] fax signal, or [[a]] data signal;

if the type of the depacketized signal is [[a]] voice signal, performing a voice mode signal processing on the ~~voice signal~~ depacketized signal;

if the type of the depacketized signal is [[a]] fax signal, performing a fax relay mode signal processing on the depacketized signal;

if the type of the depacketized signal is [[a]] data signal, performing a data modem relay mode signal processing on the depacketized signal;

transmitting the depacketized processed signal to a corresponding type of telephony device of the plurality of telephony devices; and

dynamically enabling and disabling signal processing functionality during processing of the depacketized signal, to control processor computational load.

Claim 176. (Previously Presented) The method of claim 175, wherein the plurality of telephony devices include one or more of analog and digital telephones, ethernet telephones, internet protocol telephones, analog fax machines, data modems, cable modems, interactive voice response systems, and private branch exchange systems.

Claim 177. (Previously Presented) The method of claim 175, wherein the packet based network is the internet.

Claim 178. (Currently amended) A method for integrated interfacing of a plurality of telephony devices to a packet based network, the packet based network adapted for transmission of packetized signals, the method comprising:

estimating a pitch period of ~~detecting human voice or lack thereof in~~ a voice band signal using an autocorrelation function~~based on pitch period of [[a]] the voice band signal~~;

comparing the estimated pitch period to a plurality of thresholds;

packetizing a voice signal, a fax signal, or a data signal in a packetization engine to generate a packetized signal, based upon the ~~detecting~~ comparing and at least one power measurement of the voice band signal; and

transmitting the packetized signal over the packet based network to a far end telephony device.

Claim 179. (Previously presented) A signal processing system, for interfacing telephony devices with packet-based networks, the system comprising:

a voice exchange for exchanging voice signals between a network line and a packet based network;

a full duplex data exchange for exchanging data signals from the network line with data signals from the packet based network, wherein the full duplex data exchange demodulates the data signals from the network line, outputs the demodulated data signals to the packet based network, remodulates demodulated data signals from the packet based network, and outputs the remodulated data signals to the network line;

wherein the data signals from the network line are modulated by a voiceband carrier, and the data exchange comprises a data pump for demodulating the data signals from the network line for transmission on the packet based network and remodulating the data signals from the packet based network with the voiceband carrier for transmission on the network line at a transmit rate; and

wherein the data exchange comprises a jitter buffer for receiving packets of the data signals of varying delay from the packet based network and compensating for the delay variation of the data signal packets by holding a number of the received data signals, and a clock

synchronizer which adaptively adjusts the transmit rate of the data pump in response the number of the received data signals in the jitter buffer.

Claim 180. (Previously presented) A signal processing system, for interfacing telephony devices with packet-based networks, the system comprising:

a voice exchange for exchanging voice signals between a network line and a packet based network;

a full duplex data exchange for exchanging data signals from the network line with data signals from the packet based network, wherein the full duplex data exchange demodulates the data signals from the network line, outputs the demodulated data signals to the packet based network, remodulates demodulated data signals from the packet based network, and outputs the remodulated data signals to the network line;

wherein the data signals from the network line are modulated by a voiceband carrier, and the data exchange comprises a data pump for demodulating the data signals from the network line for transmission on the packet based network and remodulating the data signals from the packet based network with the voiceband carrier for transmission on the network line at a transmit rate;

wherein the data exchange comprises a jitter buffer for receiving packets of the data signals of varying delay from the packet based network and compensating for the delay variation of the data signal packets by holding a number of the received data signals, and spoof logic which provides spoof data to the data pump when the number of the received data signals held in the jitter buffer is below a threshold.

Claim 181. (Previously presented) A signal processing system, for interfacing telephony devices with packet-based networks, the system comprising:

a voice exchange for exchanging voice signals between a network line and a packet based network, wherein the voice exchange comprises a jitter buffer for receiving packets of the voice signals of varying delay from the packet based network and compensating for the delay variation of the voice signal packets, and wherein the jitter buffer comprises a voice queue which buffers the received voice signals for a holding time, and a voice synchronizer which adaptively adjusts the holding time of the voice queue;

a full duplex data exchange for exchanging data signals from the network line with data signals from the packet based network, wherein the full duplex data exchange demodulates the data signals from the network line, outputs the demodulated data signals to the packet based network, remodulates demodulated data signals from the packet based network, and outputs the remodulated data signals to the network line; and

a tone exchange for exchanging DTMF signals between the network line and the packet based network, the DTMF exchange comprising a DTMF queue for buffering packets of the DTMF signals from the packet based network, and a tone generator which generates a DTMF tone responsive to the buffered DTMF signals, the DTMF queue outputting a signal to the voice synchronizer to suppress the buffered voice signals when the DTMF signals are in the DTMF queue.

Claim 182. (Previously presented) A signal processing system, comprising:

a voice exchange for exchanging voice signals between a first telephony device and a packet based network, wherein the voice exchange comprises a decoder for decoding packets of the voice signals from the packet based network, an echo canceller for canceling decoded voice signal echoes on incoming voice signals from the first telephony device, and a non-linear

processor which mutes the incoming voice signals when the incoming voice signals do not comprise speech and the echo canceller detects the decoded voice signals with speech; and

a full duplex data exchange for exchanging data signals from a second telephony device with demodulated data signals from the packet based network, wherein the full duplex data exchange demodulates data signals from the first telephony device, outputs the demodulated data signals to the packet based network, remodulates demodulated data signals from the packet based network, and outputs the remodulated data signals to the first telephony device; and a call discriminator for selectively enabling at least one of the voice exchange and the data exchange.

Claim 183. (Previously presented) The signal processing system of claim 1, wherein processor resources comprise one of memory, processing capacity, and power consumption.

Claim 184. (Currently amended) The signal processing system of claim 1, wherein dynamically ~~adjusting complexity of signal processing algorithms~~ <u>enabling and disabling signal processing functionality</u> comprises selecting from a plurality of levels of functionality of an algorithm.

Claim 185. (Previously presented) The signal processing system of claim 1, wherein dynamically adjusting complexity of signal processing algorithms comprises one of changing an amount of time between filter adaptations, bypassing or disabling an echo canceller, and bypassing or disabling a filter.

Claim 186. (Previously presented) The signal processing system of claim 26, wherein the estimate of pitch period of the voice band signal is calculated by applying an autocorrelation function and a plurality of power measurements to the voice band signal.

Claim 187. (Currently amended) The method of claim 49, wherein dynamically ~~adjusting complexity of signal processing algorithms~~ <u>enabling and disabling signal processing functionality</u> comprises selecting from a plurality of levels of functionality of an algorithm.

Claim 188. (Previously presented) The method of claim 49, wherein dynamically adjusting complexity of signal processing algorithms comprises one of changing an amount of time between filter adaptations, bypassing or disabling an echo canceller, and bypassing or disabling a filter.

Claim 189. (Previously presented) The method of claim 74, wherein the estimate of pitch period of the voice band signal is calculated by applying an autocorrelation function and a plurality of power measurements to the voice band signal.

Claim 190. (Previously presented) The method of claim 175, wherein dynamically ~~adjusting complexity of signal processing algorithms~~ enabling and disabling signal processing functionality comprises selecting from a plurality of levels of functionality of an algorithm.

Claim 191. (Previously presented) The method of claim 175, wherein dynamically adjusting complexity of signal processing algorithms comprises one of changing an amount of time between filter adaptations, bypassing or disabling an echo canceller, and bypassing or disabling a filter.

Claim 192. (Previously presented) The method of claim 178, wherein the estimate of pitch period of the voice band signal is calculated by applying an autocorrelation function and a plurality of power measurements to the voice band signal.